IN THE CLAIMS

Sub 7 1. (Currently Amended) A ~~gateway~~ system for throttling network packets generated from an audio signal, comprising:

an encoder that encodes the audio signal into audio packets;

an interface buffer for storing the audio packets; and

a processor that converts the audio packets into the network packets ~~,~~ and monitors available utilization capacity for at least one of the interface buffer and the processor;

the processor controlling an amount the audio signal that the encoder encodes into the audio packets according to the monitored available utilization capacity ~~of the gateway for converting additional audio packets into network packets~~.

2. (Currently Amended) A gateway ~~according to claim 1 including~~ for throttling network packets generated from an audio signal, comprising:

an encoder that encodes the audio signal into audio packets;

a processor that converts the audio packets into the network packets, the processor controlling an amount the audio signal that the encoder encodes into the audio packets according to available utilization capacity of the gateway for converting additional audio packets into network packets; and

a buffer having a free queue that receives the audio packets from the encoder, the available utilization capacity of the gateway varying according to space available in the free queue for receiving the audio packets.

3. (Original) A gateway according to claim 2 wherein the processor increases an amount of samples of the audio signal the encoder encodes into the audio packets when space available in the free queue falls below a first threshold and decreases the amount of samples of the audio signal the encoder encodes into the audio packets when space available in the free queue rises above a second threshold greater than the first threshold.

4. (Original) A gateway according to claim 3 wherein the space available in the free queue is inversely proportional with a number of network packets in the buffer waiting to be transmitted over an IP network.

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5. (Currently Amended) A ~~gateway~~ system according to claim 1 wherein the utilization capacity ~~of the gateway~~ varies according to a number of audio signals from incoming calls the ~~gateway~~ system is currently converting into network packets.

6. (Currently Amended) A ~~gateway~~ system according to claim 1 including multiple encoders each encoding audio signals into audio packets for a different incoming call, the processor varying a percentage of the encoders that increase the audio packet size according to the utilization capacity ~~of the gateway~~.

7. (Currently Amended) A ~~gateway~~ system according to claim 1 wherein the encoder encodes about 20 milliseconds of the audio signal into the audio packets when the available utilization capacity ~~of the gateway~~ is greater than a first threshold, encodes about 40 milliseconds of the audio signal into the audio packets when the available utilization capacity ~~of the gateway~~ falls below the first threshold, and encodes more than 60 milliseconds of the audio signal into the audio packets when the available utilization capacity ~~of the gateway~~ falls below a second threshold less than the first threshold.

8. (Currently Amended) A ~~gateway~~ system according to claim 1 wherein the audio signal is received over an incoming ~~PSTN~~ Public Services Telephone Network (PSTN) call and the network packets are IP packets transferred out over an IP network.

9. (Currently Amended) A method for throttling network packets in a voice gateway, comprising:

encoding an audio signal;

formatting the encoded audio signal into ~~VoIP~~ Voice Over Internet Protocol (VoIP) packets using a central processing unit;

storing the VoIP packets in an interface buffer;

monitoring utilization of at least one of the interface buffer and the central processing unit; and

controlling size of the VoIP packets by varying a number of samples of the encoded audio signal in the VoIP packets according to the monitored utilization.

10. (Original) A method according to claim 9 including formatting the encoded audio signals using the central processing unit and varying the VoIP packet size according an amount of processing capacity of the central processing unit used for formatting the encoded audio signal into VoIP packets.

11. (Original) A method according to claim 10 including:

storing the VoIP packets in the interface buffer before transmitting the VoIP packets over a VoIP network;

monitoring the interface buffer by determining an amount of free space in the interface buffer currently not storing VoIP packets; and

controlling the VoIP packet size according to the amount of free space currently in the interface buffer.

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12. (Original) A method according to claim 11 including periodically monitoring the amount of free space in the interface buffer and the available processing capacity of the central processing unit and controlling the VoIP packet size according to that periodic monitoring.

13. (Original) A method according to claim 11 including using multiple digital signal processors to encode multiple audio signals at the same time and varying a percentage of the digital signal processors that increase the VoIP packet size according to the amount of free space in the interface buffer and an amount of processing capacity of the central processing unit used for switching the encoded audio signal to the IP network.

14. (Currently Amended) A method according to claim 9 wherein formatting the encoded audio signal into VoIP packets includes the following:

attaching an ~~IP~~ Internet Protocol header to the encoded audio signal;

attaching a ~~UDP~~ User Datagram Protocol (UDP) header to the encoded audio signal; and

attaching ~~an RTP~~ a Realtime Transport Protocol (RTP) header to the encoded audio

signal.

15. (Original) A method according to claim 9 including increasing a number of samples of the audio signal in the VoIP packets when utilization in the interface buffer is above a first threshold and lowering the number of samples of the audio signal samples in the VoIP packets when utilization in the interface buffer drops below a second threshold lower than the first threshold.

16. (Original) A computer program for use with a network processing device, said computer program, comprising:

a processor load monitor that monitors utilization of a processor in the network processing device;

a buffer load monitor that monitors utilization of an interface buffer that buffers audio packets in the network processing device; and

a throttle indicator that generates a throttle value according to the monitored processor utilization and monitored interface buffer utilization, the throttle value used by the network processing device to vary an amount of an audio signal that is encoded into the audio packets.

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17. (Currently Amended) A computer program according to claim ~~17~~ 16 wherein size of the audio packets are throttled in a percentage of multiple digital signal processors wherein the percentage is proportional to the throttle value.

18. (Original) A computer program according to claim 16 wherein a number of samples of the audio signal encoded into the audio packets is increased when the monitored processor utilization reaches a first processor utilization threshold or the monitored interface buffer utilization reaches a first buffer threshold.

19. (Original) A computer program according to claim 18 wherein the number of samples of the audio signal encoded in the audio packets is decreased when the monitored processor utilization drops below a second processor utilization threshold lower than the first processor utilization threshold and the monitored interface buffer utilization drops below a second buffer threshold lower than the first buffer threshold.

20. (New) A system for throttling network packets in a voice gateway, comprising:

means for encoding an audio signal;

means for formatting the encoded audio signal into Voice over Internet Protocol (VoIP) packets using a central processing unit;

means for storing the VoIP packets in an interface buffer;

means for monitoring utilization of at least one of the interface buffer and the central processing unit; and

means for controlling size of the VoIP packets by varying a number of samples of the encoded audio signal in the VoIP packets according to the monitored utilization.

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21. (New) A system according to claim 20 including means for formatting the encoded audio signals using the central processing unit and varying the VoIP packet size according an amount of processing capacity of the central processing unit used for formatting the encoded audio signal into VoIP packets.

22. (New) A system according to claim 21 including:

means for storing the VoIP packets in the interface buffer before transmitting the VoIP packets over a VoIP network;

means for monitoring the interface buffer by determining an amount of free space in the interface buffer currently not storing VoIP packets; and

means for controlling the VoIP packet size according to the amount of free space currently in the interface buffer.

23. (New) A system according to claim 22 including means for periodically monitoring the amount of free space in the interface buffer and the available processing capacity of the central processing unit and controlling the VoIP packet size according to that periodic monitoring.

24. (New) A system according to claim 22 including means for using multiple digital signal processors to encode multiple audio signals at the same time and varying a percentage of the digital signal processors that increase the VoIP packet size according to the amount of

free space in the interface buffer and an amount of processing capacity of the central processing unit used for switching the encoded audio signal to the IP network.

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25. (New) A system according to claim 20 including:

means for attaching an Internet Protocol header to the encoded audio signal;

means for attaching a User Datagram Protocol (UDP) header to the encoded audio signal; and

means for attaching a Realtime Transport Protocol (RTP) header to the encoded audio signal.

26. (New) A system according to claim 20 including means for increasing a number of samples of the audio signal in the VoIP packets when utilization in the interface buffer is above a first threshold and lowering the number of samples of the audio signal samples in the VoIP packets when utilization in the interface buffer drops below a second threshold lower than the first threshold.